IN THE SPECIFICATION:

Page 1, immediately following the title, please insert the following:

This is the U.S. national phase of International Application No. PCT/EP03/08792 filed August 7, 2003, the entire disclosure of which is incorporated herein by reference.

BACKGROUND OF THE DISCLOSURE

Field of the Disclosure

The paragraphs beginning on page 1, line 2 have been changed as follows:

The ~~invention~~ disclosure relates to a method for determining the envelope curve of a modulated signal, for example for determination of the values for a CCDF diagram.

Related Technology

The determination of the envelope curve of a modulated signal is required in particular for determination of the CCDF (~~Complementary Cumulative Distribution Function~~ Complementary Cumulative Distribution Function) but also for other applications. The CCDF diagram indicates the probability that the signal level of the envelope curve of the ~~analysed~~ analyzed signal exceeds a specific level value. From the course of the CCDF diagram, the parameter of the crest factor inter alia can be determined, which parameter indicates the ratio of the power occurring at the maximum in the signal relative to the average power. The crest factor assists the operator of a modulated high frequency transmitter to determine the optimal modulation of the transmitter amplifiers. On the one hand, the transmitted power is intended to be as high as possible in order that the signal-to-noise ratio at the receivers is as large as possible. On the other hand, the transmitting power must not be too

large in order to avoid destruction due to short power peaks in the transmission amplifiers. If the measured CCDF course together with the course of an ideal signal is represented, conclusions can be made with respect to non-linearity and limitation effects in the transmitted signal.

On page 2, after line 10 please insert a heading as follows:

SUMMARY OF THE DISCLOSURE

The paragraphs beginning on page 2, line 11 have been changed as follows:

The ~~object therefore underlying the invention is to indicate~~ disclosure provides a method for determining the envelope curve of a modulated signal which operates with relatively high precision.

~~The object is achieved by the features of claim 1.~~ According to the disclosed method, the envelope curve of a modulated input signal is determined by:

- generating digital samples by digital sampling a modulated input signal ~~(S)~~,

- generating Fourier-transformed samples by Fourier transforming the digital samples,

- generating sideband-cleaned, Fourier-transformed samples by removing a range with negative frequencies or a range with positive frequencies from the Fourier-transformed samples,

- generating inverse-transformed samples by inverse Fourier transforming the sideband-cleaned, Fourier-transformed samples and

- forming values of the absolute value of the inverse-transformed samples.

In contrast to the known method, determination of the envelope curve is effected ~~according to the invention~~ not by low-pass filtering but instead the digital

samples are Fourier-transformed in the frequency range. In the frequency range, the range of positive frequencies or the range of negative frequencies is then removed. Then a Fourier inverse transform in the time domain follows. Only then are the values of the inverse-transformed samples formed. ~~It is also shown later in this application that the~~ The absolute value of the inverse-transformed samples represents the envelope curve of the modulated high frequency signal.

In contrast to the value formation and subsequent low-pass filtering, the disclosed method ~~according to the invention~~ has the advantage that implementation of the method is independent of the quality of the low-pass filtering, is independent of the type of signal and of its spectral position, and in addition independent of the ~~synchronisation~~ synchronization state of the high frequency signal to be measured. The disclosed method ~~according to the invention~~ is in addition substantially more precise than the known method with low-pass filtering.

~~The sub-claims relate to advantageous developments of the invention.~~

It is advantageous, in addition to the range of negative or positive frequencies, also to remove the level component at the DC frequency ~~0~~ zero after the Fourier transform in the frequency range. It is ensured as a result that the direct voltage offset of a non-ideal ~~analogue~~ analog/digital converter has no influence on the disclosed method ~~according to the invention~~. The ideal signal has no direct voltage component in the intermediate frequency plane so that removal of the direct voltage component does not falsify the measurement result.

The paragraph beginning on page 3, line 18 have been changed as follows:

~~Claims 6, 7, 8 and 9 relate to~~ Also disclosed are a corresponding digital storage medium, computer ~~programme~~ program or computer ~~programme~~ program product based on the disclosed method ~~according to the invention~~.

BRIEF DESCRIPTION OF THE DRAWINGS

~~The invention is described in more detail subsequently with reference to the drawing. There are shown in the drawing:~~

Fig. 1 shows an example of a CCDF diagram;

Fig. 2 shows a block diagram of the disclosed method ~~according to the invention~~;

Fig. 3 shows a diagram to explain the mode of operation of the disclosed method ~~according to the invention~~;

Fig. 4 shows the samples which are Fourier-transformed in the frequency range and

Fig. 5 shows the samples which are inverse-transformed in the time domain.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The disclosed method ~~according to the invention~~ is explained ~~subsequently~~ below for the application example for determining the instantaneous power of the envelope curve for a CCDF diagram. As already explained, the method ~~according to the invention~~ is ~~however~~ not restricted to this application and is suitable for all applications in which the instantaneous level of the envelope curve or signal values derived from the latter, such as e.g. the power, i.e. the square of the level, are required.

Fig. 2 demonstrates the method ~~according to the invention~~ by means of a block diagram. The high-frequency input signal S, which is modulated by a modulation signal, is firstly sampled digitally on a sampling and holding circuit 1. Digital

samples A_n of the input signal S are thereby produced. The samples A_n are then subjected to a Fourier transform for example with an algorithm of the fast Fourier transform (FFT, ~~Fast Fourier Transform~~ Fast Fourier Transform). The Fourier-transformed samples B_n are produced as a result. The Fourier transform is illustrated in Fig. 2 by block 2.

Due to the Fourier transform of a sampled real signal, Fourier-transformed samples are produced as is known, which samples extend both over the range of negative frequencies and over the range of positive frequencies. According to the ~~invention~~ method, either the range of negative frequencies or the range of positive frequencies is removed from the Fourier-transformed samples B_n. If the index n is running, which indexes the Fourier-transformed samples B_n, for example from $-2^N/2$ to $2^N/2-1$, N being a whole natural number, then the range of negative frequencies corresponds to the samples B_n with $n < 0$ or the range of positive frequencies corresponds to the samples B_n with $n > 0$.

The remaining samples, which are either only positive or only negative, are designated in Fig. 2 with B'_n. Trimming of the samples in the negative frequency range is illustrated in Fig. 2 by the block 3 which has a transfer function H(f) which is different from 0 only in the range of positive frequencies. These sideband-cleaned, Fourier-transformed samples B'_n are subsequently transformed back in the time domain by an inverse Fourier transform. Likewise, a fast digital Fourier inverse transform (IFFT, ~~Inverse Fast Fourier Transform~~ Inverse Fast Fourier Transform) can be used, which is illustrated in Fig. 2 by block 4. In the time domain, the inverse-transformed samples C_n are present, the value of which is still to be formed finally in the value former 5. The value of the samples, which are inverse-transformed in the time domain, is designated in Fig. 2 with D_m.

The paragraph beginning on page 5, line 18 has been changed as follows:

In the case of application of the CCDF diagram, there must now be established in a block 6 the relative frequency with which the square of the value-samples D^2_m, which corresponds to the power, exceeds a threshold value x in relation to the average power D^2_{eff} on a logarithmic scale which is scaled in dB. Expediently, the squaring is implemented not before but after ~~logarithmising~~ logarithmizing, i.e. instead of a multiplication by the factor 10, a multiplication by the scaling factor 20 is effected:

The paragraph beginning on page 7, line 8 has been change as follows:

The function of the disclosed method ~~according to the invention~~ is described in more detail with reference to Figs. 3 and 4. The signal S can be ~~factorised~~ factorized in a Fourier sequence, i.e. any arbitrary input signal can be constructed from a series of cosine signals with different signal levels and phases. In the following, only one of these Fourier components is considered, which can be written in general as follows:

The paragraph beginning on page 8, line 1 has been changed as follows:

The signal s_1 (t) comprises a first rotating vector 8, which rotates to the left with the angle frequency ù, and a second rotating vector 9 ~~synchronised~~ synchronized thereto which rotates to the right with the same circular frequency ù. The omission of the range of negative frequencies ~~according to the invention~~ leads to the fact that the rotating vector 9 is suppressed. In reverse, omission of the range of positive frequencies, which is just as possible as an alternative, leads to the fact that the rotating vector 8 is suppressed. Filtering in the frequency range leads therefore to

omission of one of the two terms in equation (5). If for example the component with the negative frequency, i.e. the rotating vector 9 which rotates to the left in Fig. 3, is omitted in equation (4), then the following result is produced after the amount formation:

The paragraphs beginning on page 9, line 11 have been changed as follows:

Expediently, not only either the range 10 of negative frequencies or the range 11 of positive frequencies is suppressed, but in addition also the level component 12 for the zero frequency; in the indexation used here, i.e. B_0 with $n = 0$. Thus a possibly present direct voltage component (DC-offset) is suppressed. Since the evaluated signals stem from the intermediate frequency plane, these should actually contain no direct voltage component. If however a direct voltage component is present, then this stems for example from a direct voltage offset of the ~~analogue~~ analog-digital converter and removal of this direct voltage component increases the measurement precision.

An example of a CCDF diagram, the underlying envelope curve of which was obtained with the disclosed method ~~according to the invention~~, is illustrated in Fig. 1. As already mentioned, the relative frequency p is plotted for this purpose in a CCDF diagram such that a specific level D on a logarithmic scale is exceeded. In the example illustrated in Fig. 3 of an input signal which has been modulated digitally according to the 8VSB standard, exceeding the effective power with 3 dB occurs still with a relative frequency of approximately 10%, whilst exceeding the effective power with more than 6 dB occurs already with a relative frequency significantly smaller than 1%.

As already mentioned ~~many times~~, the disclosed method ~~according to the invention~~ is not restricted to the application case for determining instantaneous level values or instantaneous power values for a CCDF diagram, but in general is suitable for determining the envelope curve of a modulated signal. The method can be implemented both with digital hardware, for example by using FPGA (~~Free Programmable Gate Array~~ Free Programmable Gate Array), or with software in a special processor, ideally in a digital signal processor (DSP).